ALCATEL LUCENT MEDIA GATEWAY

7515 R31S Release Notes

7515 Release R31s (S31B0110.pak)

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1 Build Description and Name

The build name for this software delivery is

- A7515S31(S31B0110.pak)

Files included in this package:
\\ngnsvr2\SLP\7515S3.1\7515S3.1\7515S_R31_08_07_2009(*S31B0110)*

- S31B0110.pak
- AlcatelTG_v7_17_2_1.axf
- config.txt
- GW_Release_Notes_7515S31.doc

2 References

[1] [3FZ045565006ACZZA] 7515R3.1S Operation Guide
[2] [3FZ045565002ACZZA] 7515R3.1S User Manual
[3] [3FZ045565005ACZZA] 7515R3.1S Quick Start
[4] [3FZ045565004ACZZA] a751531S Production Description

3 Features

3.1 Features Released in 3.1S

3.1.1 Basic Call per TS183036 Version 0.10.1

New Features introduced to build:

- Basic call feature (xml mapping) per TS183036 Version 0.10.1 (2007-11)
- SIP reason header support as per rfc3326
- P-early-media header support per rfc5009

3.1.1.1 Basic call (To-NET)

3.1.1.1.1 Q.931 Setup -> SIP Invite

- Bearer capability (BC) IE mapping to BC XML and SDP
- HLC, LLC, PI mapping to XML
- Called Party Number IE mapping to Request-URI and To
- Coding of SDP from BC/HLC
- P-Early-Media header

3.1.1.1.2 SIP 180 Ringing -> Q.931 Alerting

- PI XML attachment and P-Early-Media header mapping to PI IE

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- HLC XML attachment mapping to HLC IE
- Handling of fallback information
    - BC fallback
    - HLC fallback
    - SDP fallback (7KHZ audio fallback to 3.1kHZ audio)

3.1.1.1.3 SIP 183 Session Progress -> Q.931 Call Proceeding or Progress

- PI XML attachment and P-Early-Media header mapping to PI IE
- HLC XML attachment mapping to HLC IE
- Handling of fallback information
    - HLC fallback

3.1.1.1.4 SIP 200 OK -> Q.931 Connect

- Process of “from-change” tag
- PI XML mapping to PI IE
- HLC/LLC XML attachment mapping to HLC/LLC IE
- Handling of fallback information
    - BC fallback
    - HLC fallback

3.1.1.1.5 SIP Bye (or final Response) -> Q.931 Disconnect

- Reason header mapping to cause IE
- PI setting in Q.931 Disconnect
- Q.931 Disconnect/Release/Release Complete -> SIP Bye/Cancel
- Cause IE -> Reason header in SIP Bye (or Cancel)

3.1.1.2 Basic call (To-TELE)

3.1.1.2.1 SIP Invite -> Q.931 Setup

- Support of 606 status code
- Bearer capability (BC) mapping
- Support of 415 status code in case b= line indicate a bandwidth > 64kbit/s.
- Mapping P-Asserted-ID/From/Privacy to Calling Party Number
- Request-URI mapping to Called Party Number

3.1.1.2.2 Q.931 Call Proceeding / Progress -> SIP 183

- PI IE mapping to PI XML
- HLC, LLC IE mapping to HLC, LLC XML
- Inserting of “P-Early-Media” header into 183
- History-Info header
- P-Notification header
- Handling of fallback information

3.1.1.2.3 Q.931 Alerting -> SIP 180

- PI IE mapping to PI XML
- Inserting of “P-Early-Media” header into 180

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- History-Info header
- P-Notification header
- Handling of fallback information

3.1.1.2.4 Q.931 Connect -> SIP 200

- P-Asserted-Identity
- PI, LLC IE mapping to PI, LLC XML
- Handling of fallback information (BC and HLC)

3.1.1.2.5 SIP Bye/Cancel -> Q.931 Disconnect

- Reason Header in SIP Bye/Cancel -> Cause IE in Q.931 Disconnect

3.1.1.2.6 Q.931 Disconnect/Release/Release Complete -> SIP

- Cause IE -> Reason header in SIP Bye/4xx/6xx

3.1.1.3 New UI Commands introduced:

define voip sip xml-support

Syntax
define voip sip xml-support {enable/disable}
Description
enable or disable whether to insert xml in outgoing SIP Messages or decode xml in incoming SIP message.

view voip sip xml-support

Syntax
view voip sip xml-support
Description
Show the state of whether to insert or decode xml in SIP Message body.

3.1.1.4 Old UI Commands removed:

define voip sip early-media

view voip sip early-media

3.1.2 Supplementary Services feature (xml mapping) per TS183036 V0.15.1

New Features introduced to build:

Connected Line Identification Presentation (COLP)
Connected Line Identification Restriction (COLR)
Connecting Line Identification Presentation (COLP)
Connecting Line Identification Restriction (COLR)

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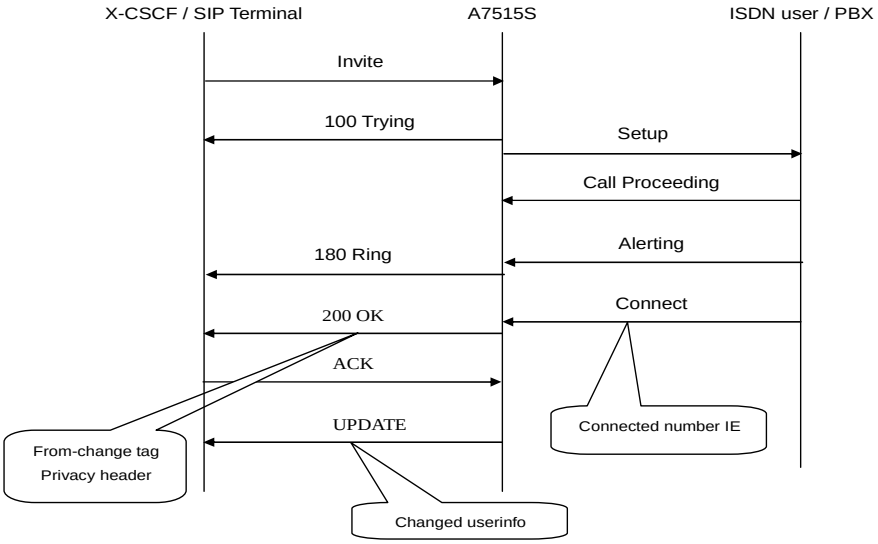
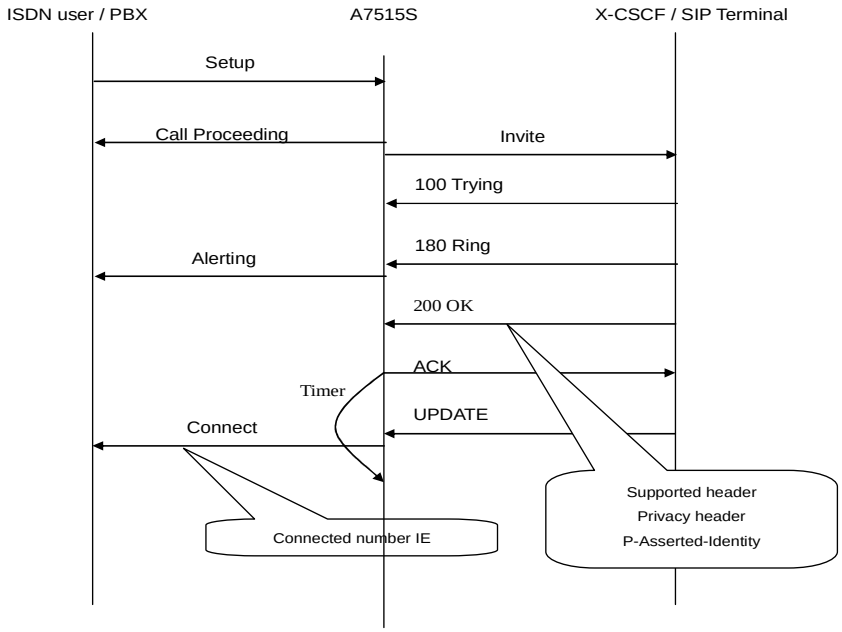
3.1.2.1 To-TELE COLP/COLR call establishment



3.1.2.2 To-NET COLP/COLR call establishment(receive UPDATE message)



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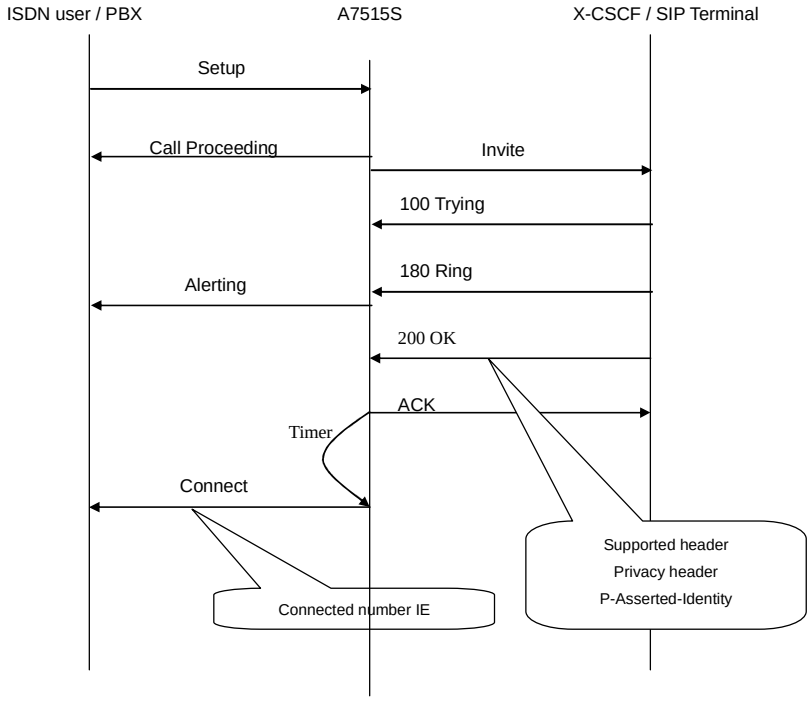
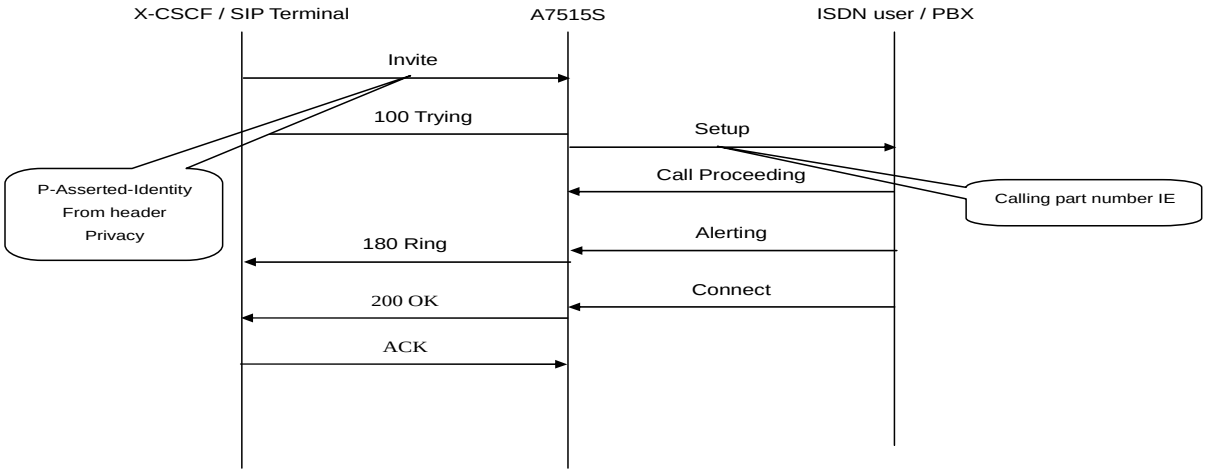
3.1.2.3 To-NET COLP/COLR call establishment(Timer expire)



3.1.2.4 To-TELE CLIP/CLIR call establishment



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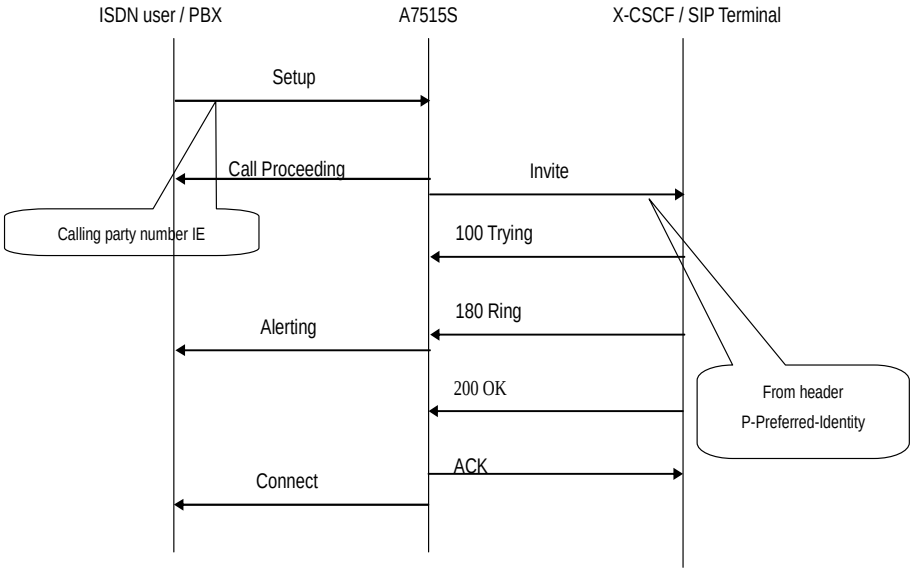
3.1.2.5 To-NET CLIP/CLIR call establishment



3.1.2.6 New UI Commands introduced:

define voip sip colp Tupdate

Description

Specifies the colp Tupdate Timer interval.

Arguments

<val> the colp Tupdate Timer interval (ms).

Range: 3,000-10,000 ms

view voip sip colp config

Description

Shows the sip colp *feature configuration.*

define voip sip interface type

Description

Specifies sip interface is gm interface or mw interface

Arguments

{gm| mw}: gm or mw, only one of them, default value is gm.

view voip sip interface

Description

Shows the interface type: Gm or Mw

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3.1.3 Register Feature overview

New Features introduced to build:
SIP Register feature, based on TRS 8CH01987AHAA_DTZZA, RFC 3261, and ETSI TS 123 228 V8.5.0 (2008-06)
Trunk group selection for to-TELE calls, as described in 3FZ-045561330-DSZZA

- The SIP Register feature consists of adding SIP registration support to the A7515 gateway on a per trunk group basis. Each trunk group is composed of at least one link (E1/T1) and is assigned a public user identity (PUID) and, optionally, a private user identity (PRID) and password. In case the PRID and password are provided, the gateway is able to work with authentication, if required by the server.
-
- If the trunk group is configured to use registration, as soon as at least one of its links gets in-service, the registration procedure is started.
-
- A list of associated URIs can be assigned to a given trunk group, either via UI command or when the 200 OK response to a REGISTER request contains a P-Associated-URI header. This list can be used to perform the trunk group selection for to-TELE calls and can also be used to perform calling party's number screening.
-

3.1.3.1 New UI Commands introduced:

define voip sip register id
Syntax:
define voip sip register id <trunk-group> <puid> [prid] [pwd]
Description:
This command defines the user identification and credentials to be used when registering a given trunk group. In order to be registered, the trunk group state must have been set to 'enable-reg'.
Arguments:
<trunk-group>
The valid values range from 1 to 480.
<puid>
The Public User Identity associated with the trunk group, which usually corresponds to the PBX main number. When the trunk group is registered, a set of other public identities can be implicitly registered so the PBX's extensions can be known within the IMS network.
<prid>
The Private User Identity is used for registration, authorization, administration and accounting purposes. In case this parameter is not provided, authentication will not be supported during registration and, if challenged by the registrar, the registration will fail.

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<pwd>
The password is used in conjunction with the PRID as the user's credentials, for authentication purposes.

Business Rules & Dependencies:
the puid and prid must be unique, i.e. it must not have been previously defined in another trunk group.

Availability:
This command is only supported on the active FSB board.

Example:
define voip sip register id 3 +553130694000 +553130694000 TrkGrp_1-PaSsWoRd

define voip sip register associated

Syntax:
define voip sip register associated <trunk-group> <puid> [default]

Description:
This command adds a PUID to the P-Associated-URI list assigned to a given trunk group. Wildcard PUIDs such as "tel:+16302241![0-9][0-9][0-9]!" should be accepted. The values provisioned may be used for trunk group selection (to-TELE calls) and calling party's number screening (to-NET calls). In case the trunk group requires registration and the 200 OK response received from the registrar contains a P-Associated-URI header, the list provided by the registrar will be used instead (the list provisioned by the craft is kept and can be queried but will not be used).

Arguments:
<trunk-group>
The trunk group the list is associated with.

<puid>
The PUID to be added to the list.

[default]
When provided, this parameter set the PUID as the trunk group default PUID. If there is no default PUID assigned to the trunk group, the first PUID added to the list is considered the default one.

Business Rules & Dependencies:
- the trunk group must have been previously configured with the 'define voip sip register id' command;

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- the puid must not have been previously added to another associated URI list, even if it is assigned to a different trunk group; special care must be taken in case wildcard URIs are used.

Availability:
This command is only supported on the active FSB board.

Example:
define voip sip register associated 3 +553130694001 default

remove voip sip register associated

Syntax:
remove voip sip register associated <trunk-group> <puid>

Description:
This command deletes a PUID from the P-Associated-URI list assigned to a given trunk group. In case the PUID provided is the default PUID, the first entry in the list will be used as the default number until another PUID is provisioned as default.

Arguments:
<trunk-group>
The trunk group the list is associated with.

<puid>
The PUID to be deleted from the list.

Business Rules & Dependencies:
the trunk group must have been previously configured with the ‘define voip sip register id’ command.

Availability:
This command is only supported on the active FSB board.

Example:
remove voip sip register associated 3 +553130694001

define voip sip register state

Syntax:
define voip sip register state {trunk-group|all} {disable| enable|enable-reg}

Description:
This command defines the trunk group maintenance state.

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Arguments:
{trunk-group|all}
The trunk group that will have its state changed. In case ‘all’ is specified, the state change will affect all trunk groups.

{disable|enable|enable-reg}
The new maintenance state, which can be as follows:
- disable: disables the trunk group. When the trunk group is in this state it is out of service and no calls will be routed through it. In case it was previously registered, it must be de-registered
- enable: enables the trunk group, no registration is required. The trunk group will be in service when its bearer channels are up
- enable-reg: enables the trunk group, registration is required. The trunk group will be in service in case its bearer channels are up and the registration has succeeded.

Business Rules & Dependencies:
the trunk group must have been previously configured with the ‘define voip sip register id’ command.

Availability:
This command is only supported on the active FSB board.

Example
define voip sip register state all enable-reg

define voip sip register domain-name

Syntax:
define voip sip register domain-name <home-domain-name>

Description:
This command defines the domain name to be used when registering the A7515 subscribers. When defined, this domain name is sent in the Request URI line of a REGISTER request and is also used in the address of record sent in both the To and From headers. As an example, consider the domain name ‘example.com’. The REGISTER request will be populated as follows:

REGISTER sip:example.com SIP/2.0
From: <sip:+551137077801@example.com;user=phone>;tag=22134
To: <sip:+551137077801@example.com;user=phone>

Arguments:
<home-domain-name>
The domain name to be used when registering the local subscribers.

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Business Rules & Dependencies:
- In case the multicast address is used when registering the subscribers, the domain name will not be used in the Request URI line. The request sent will be like the following:

REGISTER sip:224.0.1.75 SIP/2.0
From: <sip:+551137077801@example.com;user=phone>;tag=22134
To: <sip:+551137077801@example.com;user=phone>

- a DNS server must be configured in order to resolve the domain name provided, otherwise the REGISTER request will fail.

Availability:
This command is only supported on the active FSB board.

Example
define voip sip register domain-name “example.com”

define voip sip register host-name

Syntax:
define voip sip register host-name <hostname>

Description:
This command defines the host name to be used when registering the local subscribers. In case a domain name is configured for the A7515 (which is the domain name corresponding to the gateway actual location and is not the same as the one configured for the register), the host name will be used in the Contact header of the REGISTER request. The gateway domain name should be defined using the existing UI command ‘define ip dns domain’.

As an example, if the register domain name is ‘example.com’, the domain where the A7515 is located is ‘some_carrier.com’ and the host name is ‘br_gw421’, the following request will be sent:

REGISTER sip:example.com SIP/2.0
From: <sip:+551137077801@example.com;user=phone>;tag=22134
To: <sip:+551137077801@example.com;user=phone>
Contact: <sip:+551137077801@br_gw421.some_carrier.com;user=phone>

In case either the host name or the gateway domain name are not configured, the gateway IP address will be used in the Contact header instead:

REGISTER sip:example.com SIP/2.0
From: <sip:+551137077801@example.com;user=phone>;tag=22134
To: <sip:+551137077801@example.com;user=phone>

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Contact: <sip:+551137077801@135.245.10.231;user=phone>

Arguments:
<hostname>
The host name to be used when forming the Contact header.

Business Rules & Dependencies:

Availability:
This command is only supported on the active FSB board.

Example
define voip sip register host-name “br_gw421”

view voip sip register config

Syntax:
view voip sip register config {trunk-group|all}

Description:
This command displays the trunk group configuration, presenting the following data:
- trunk group maintenance state (disable, enable or enable-reg);
- trunk group operational state, which is composed of a primary state (in service or out of service) and a secondary state, which is optional and is used to provide additional information such as link down, registering, registered, not registered, registration failed, etc.);
- the PUID and PRID associated with the trunk group
- the P-Associated-URI lists associated with the trunk group (the provisioned and the active, which may be different in case registration is enabled and the registrar response contains a P-Associated-URI header).

Arguments:
{trunk-group|all}
This parameter determines the trunk group whose configuration will be displayed. In case ‘all’ is specified, the configuration of all provisioned trunk groups is displayed.

Business Rules & Dependencies:
the trunk group must have been previously configured with the ‘define voip sip register id’ command.

Availability:
This command is supported on both the active and the standby FSB boards.

Example:

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view voip sip register config 1

SIP Gateway State : enabled
SIP Register Domain Name : example.com
SIP Register Host Name : gw12
SIP Register Auto Discover : disabled
SIP Register Max Times : 2
SIP Register Expire Time : 60000
SIP Register Refresh Time : 59950
Sip Registrar Primary Ip Address : 10.31.1.126
Sip Registrar Primary Ip Port : 5062
Sip Registrar Primary Scheme : UDP
Sip Registrar Secondary Ip Address : 10.31.1.127
Sip Registrar Secondary Ip Port : 5060
Sip Registrar Secondary Scheme : UDP
Sip Registrar Active Ip Address : 10.31.1.126
Sip Registrar Active Ip Port : 5062
Sip Registrar Active Scheme : UDP

****
Trunk Group 1 :

Maintenance State: disable
Operational State: OOS-OOS
PUID : +553130694000
PRID : +553130694000

Active Associated URI List:

+553130694![0-1][0-9][0-9]! (default)

Provisioned Associated URI List:

+553130694000 (default)
+5531306941![0-9][0-9]!

define int-num-prefix

Syntax:
define int-num-prefix <trunk-group> <country-code> <area-code>

Description:
This command defines the international number prefix assigned to the trunk group.

Arguments:
<trunk-group>
The trunk group identification.

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<country-code>
The country code to be assigned to the trunk group

<area-code>
The area code to be assigned to the trunk group

Business Rules & Dependencies:

Availability:
This command is only supported on the active FSB board.

Example
define int-num-prefix 5 1 630

view international-number-prefix

Syntax:
view int-num-prefix {trunk-group|all}

Description:
This command displays the country and area codes assigned to a trunk group.

Arguments:
{trunk-group|all}
The trunk group identification. In case ‘all’ is specified, the international number prefix for all trunk groups is displayed.

Business Rules & Dependencies:

Availability:
This command is supported on both the active and the standby FSB boards.

Example
view int-num-prefix all

3.1.3.2 **Removed UI Commands:**

define voip sip alias
view voip sip alias
remove voip sip alias
view voip sip register contacts

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3.1.4 Dynamic Payload type /CLEARMODE Feature overview

New Features introduced to build:

- Dynamic payload type feature (Clear mode and DTMF NTE relay over RTP)

3.1.4.1 Clearmode

3.1.4.1.1 To-Net: Setup with UDI

If the UDI is indicated in bearer capability IE of SETUP, it is to say that the media should be adopt Clearmode. If Clearmode is contained in profile, then all other codecs except Clearmode should be removed from SDP. Namely, only Clearmode is included in SDP.

3.1.4.1.2 To-Net: Setup with UDI, but no Clearmode in Profile

If the UDI is indicated in bearer capability of SETUP, it is to say that the media should be adopt Clearmode. If Clearmode is not contained in profile, then the call will be rejected and warning should be given.

3.1.4.1.3 To-Net: Setup without UDI, Clearmode in Profile

If UDI doesn't appear in bearer capability of SETUP, it is to say that the media should be speech not Clearmode. Even if Clearmode is contained in profile, "CLEARMODE" should not be contained in INVITE.

3.1.4.1.4 To-Tele: Clearmode is configured into Profile

The only codec in SDP of INVITE is Clearmode. If the Clearmode is configured into profile, then the call sets up with Clearmode.

3.1.4.1.5 To-Tele: Clearmode isn't configured into Profile

The only codec in SDP of INVITE is Clearmode. If the Clearmode isn't configured into profile, then the call will be rejected.

3.1.4.1.6 SDP Negotiation

If there is one codec that offer and answer both intend to, the SDP negotiation is succeeded. The codec will be used to transfer media. If there is no codec to be match, the call will fail and the answer will send 488 or other error code to offer.

3.1.4.2 Dynamic Payload Type

3.1.4.2.1 7515 As Offer

If the payload type is intended by both offer and answer, the SDP negotiation is succeeded. The payload type is used to transfer media. If the payload type in SDP

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answer is not equal to what the A7515R3.1S prefers to, A7515R3.1S will adopt the payload type from answer in order to prevent call failure.

3.1.4.2.2 7515 As Answer

A7515R3.1S as answer receives SDP offer. If the payload type in A7515R3.1S is not equal to what the offer prefers to, A7515R3.1S will adopt the payload type from offer in order to prevent call failure.

3.1.4.3 Telephone Event

There are three cases as follows:
Case 1: offer and answer both prefer to telephone event.
SDP negotiation is success. Both offer and answer supported telephone event.

Case 2: Offer prefers to telephone event, but answer doesn't support telephone event.
SDP negotiation is success. But SDP from answer doesn't contain telephone event. So telephone event isn't supported in current call.

Case 3: Offer doesn't support telephone event, but answer prefer to support telephone event.
Normally, if the offer doesn't include telephone event, the answer should not contain telephone event. Once the abnormal occurred, the offer should ignore the telephone event from answer and the call can setup successfully. So telephone event isn't supported in current call.

3.1.4.4 New UI Commands introduced:

define voip profile network payload <id> {clearmode| telephone-event } <payload_type>

define voip profile network cng <id> {g711|g7231|g729a} {enable|disable}

define voip profile network vad <id> {g711|g7231|g729a } {enable|disable}

define voip profile network digit-relay <id> sip-info {enable|disable}

define voip profile network fax control-mode <id> {softswitch|autoswitch}

define voip profile network fax autoswitch mode <id> {passthrough|t38}

define voip profile network fax para rate-mgmt <id> {local-tcf|transferred-tcf}

define voip profile network fax para udpec <id> {redundancy|FEC}

define voip profile network fax para fillBitRm <id> {enable|disable}

define voip profile network fax para ecm <id> {enable|disable}

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define voip profile network fax para pktloss <id>
{disable|whiteline|goodline}

define voip profile network fax para max-bit-rate <id>
{2400|4800|7200|9600|12000|14400}

define voip profile network fax para max-datagram <id> <value>

define voip profile network udi <id> {udi-wo-ta | udi-w-ta}

3.1.4.5 Old UI Commands updated:

define voip profile network codec <id>
{g711a|g711u|g7231|g729a|clearmode|telephone-event} {enable|disable}

define voip profile network relative codec preference <id>
{g711|g7231|g729a|clearmode} <relative-preference>

define voip profile network ptime <id> {g711|g7231|g729a|clearmode}
<ptime>

define voip profile network gain output <id> <dB>
define voip profile network gain input <id> <dB>

view voip profile updated:

vi voip pro net 2

Network Profile Data Base Entry> Id: 2 State: Ready

Codecs : g.711(A-law) g.711(u-law) g.729a g.723.1
Companding Law : A-law
Digit Relay : via voice
Loopback Mode : Off
Early B-Channel connection : disabled
Bearer Capability UDI : Unrestricted Digital Info without T/A
Echo Canceler : disabled
Echo Canceler Tail Length : 48 ms
Gain from packet to pcm : 0 dB
Gain from pcm to packet : 0 dB

Press q followed by Enter to Abort or just the Enter key to Continue:
---
g.711a/g.711u specific parameters

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Relative Preference Level : 1
Packet Time : 20 msecs
Comfort Noise Generator : enabled
Voice Activity Detection : enabled
---
g723.1 specific parameters

Relative Preference Level : 2
Packet Time : 30 msecs
Comfort Noise Generator : enabled
Voice Activity Detection : enabled
---
g.729/g.729a specific parameters

Relative Preference Level : 3
Packet Time : 30 msecs
Comfort Noise Generator : enabled
Voice Activity Detection : enabled

Parameters of fax application

FAX control mode : Soft Switch
FAX softswitch mode : Passthrough
FAX autoswitch mode : T38
Autoswitch to PassThrough codec : PCM-A
FAX softswitch T38 Voice Recover : disabled
FAX INIT EVENT : V21 event

T38 fax settings

FAX rate management : Local TCF
FAX UDP error recovery method : Redundancy
FAX Fill Bit Removal : disabled
Error Correction Mode : enabled
Packet loss concealment : disabled
FAX maximum bit Rate : 14400
FAX maximum datagram : 0
T30 udptl redundancy : 3
T4 udptl redundancy : 0
T38 media version : 0
T38 info timer : 2 seconds
Transcoding JBIG : disabled
Transcoding MMR : disabled

view voip default updated:

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vi voip def

Media Gateway default parameters :

Channel Initial Mode..... Static
Static Initial Channel Number..... 240
DIM timeout..... 5000 ms
Codec Type..... G711 a-law
Packet Time..... 30 ms
PCM law..... A-Law
CNG Mode..... Spectral
DSP Mode..... Packet Voice

Jitter Maximum Buffer Size..... 100 ms
Jitter Minimum Buffer Size..... 0
Jitter Target..... 0

COT Monitor Frequence..... 1780&2010Hz
COT Transmit Frequence..... 2010Hz
COT Active Timer..... 5000 ms

Telephone Tone transmit inRTP2833..... Enabled
DTMF inRTP2833 redundancy scheme..... AAL(RFC2833)
DTMF Removal Delay..... add extra delay
RFC2833 tone regen2TDM..... Enabled
DTMF Detection Threshold..... -35.0 db
DTMF Detection SNR Threshold..... 6 db
DTMF Redundancy Event Payload Type..... 96

VBD schema payload type..... PAYLOAD_TYPE_G711A(8)
VBD support..... Enabled
Inherit While VBD disabled..... Enabled
VBD Echo Cancellation..... Enabled

RTCP Support..... Disabled
Remote RTCP Statistics Interval..... 10 s
Remote RTCP Timeframes..... 3
Local DSP Statistics Interval..... 10 s
Local DSP Timeframes..... 3
Network Failure Mode..... 0
Disabled
FSK onhook settings
FSK onhook modulation type Bell202 (0)
FSK onhook transmit phase ..... During the ring (0)
FSK onhook alert info before FSK ..... No (0)

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FSK offhook settings
FSK offhook modulation type Bell202 (0)
FSK offhook alert info before FSK ..... No (0)
FSK cpeack settings
FSK cpeack start DTMF range ..... A~D
FSK cpeack valid DTMF duration 30
FSK cpeack valid DTMF level 300
FSK cpeack timeout 200
FSK cpeack time before send FSK 160
FSK cpeack send FSK if w/o ack ..... No (0)

3.1.5 R2 Feature

New Features introduced to build
R2 Feature (R2 to SIP VGW for Brazilian market)

Following basic call flows are supported :

- To-NET Call Establishment
- To-NET Call Disconnection
- To-TELE Call Establishment
- To-TELE Call Disconnection

This feature is the implementation of an R2 to SIP gateway prepared for the Brazilian market. The feature is basically being required for two applications: interconnecting legacy PBX and Class 5 switches via R2 to the new NGN network.

In order to support PBX interconnection the 7515S will have to be able to discriminate the trunk groups connected to PBX systems as corporate trunk groups, perform calling party screening to avoid fraud and assert the number received for the rest of the network. Trunks marked as regular, i.e., the ones that connect to the class 5 switches will not require calling party screening, since the calling party number will be received from a switch within the operator's network.

For the time being only en bloc signaling will be supported in the SIP network and so digit map will be required in the 7515S in order to know when to start the signaling exchange in the SIP side. Dialed digits length will be defined based on the leading digits, allowing the system to decide when the complete called party number had been received. For a few cases, like international calls for instance, since called party number length may vary depending on the destination, the system will implement a reduced inter-digit timer to support in the decision of when to trigger the SIP signaling and avoid the long delay that would be inserted by the standard inter-digit timer.

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3.1.5.1 New UI Commands introduced:

3.1.5.1.1 define trunk group

Description
Assign a trunk range to a trunk group

Syntax
define trunk group <first_trunk> <last_trunk> <trunk-group-id>

Arguments
<first_trunk> First trunk of range. Format {e1|t1}.scm.<port>.<channel>
<port> E1 identification, values from 1 to 16.
<channel> Time slot identification, values from 1 to 31. Value 16 is not valid since it is used for signaling.
<last_trunk> Last trunk of range. Format {e1|t1}.scm.<port>.<channel>
<port> E1 identification, values from 1 to 16.
<channel> Time slot identification, values from 1 to 31. Value 16 is not valid since it is used for signaling.
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
Availability
This command is only supported on the active FSB board.

Example
define trunk group e1.scm.1.1 e1.scm.1.31 1

3.1.5.1.2 remove trunk group

Description
Remove trunk range from trunk group or remove trunk group.

Syntax
remove trunk group {trunk-group|range}[trunk-group-id-or-first-trunk][last-trunk]

Arguments
{trunk-group|range}:
trunk-group: trunk group will be removed if there is no trunk in it.
range: all provisioned trunks within the trunk range will be removed.

[trunk-group-id-or-first-trunk]:
The trunk group id since {trunk-group|range} is trunk-group
The first trunk of the range since {trunk-group|range} is range. Format {e1|t1}.scm.<port>.<channel>. <port> E1 identification, values from 1 to 16.
<channel> Time slot identification.
(In case of channel equals 0 (ex. e1.scm.9.0) the whole link is assumed)

[last-trunk]:

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the last trunk of the range since {trunk-group|range} is range. Format {e1|t1}.scm.<port>.<channel>. <port> E1 identification, values from 1 to 16. <channel> Time slot identification.
(case this parameter was ommited, last-trunk is assumed equals first-trunk)
(In case of channel equals 0 (ex. e1.scm.9.0) the whole link is assumed)

Availability
This command is only supported on the active FSB board.

Example
remove trunk group range e1.scm.1.1 e1.scm.2.31
remove trunk group range e1.scm.1.1
remove trunk group range e1.scm.1.0
remove trunk group trunk-group 15

3.1.5.1.3 Define Trunk Signaling

Description
This command can define trunk signaling for trunk group.

Syntax
define trunk signaling <trunk_group_id> {isdn|r2|e&m}

Arguments
trunk-group-id Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
isdn|r2|e&m The desired signaling
Note: This command is only supported on the active FSB board.

3.1.5.1.4 Define Trunk Config Allocation Method

Description
This command specifies the trunk config allocation method.

Syntax
define trunk config <trunk-group-id>{cyclical|increasing|decreasing}

Arguments
trunk-group-id Specifies trunk group ID, ranging from 1 to 480 (ex: 1)

{cyclical|increasing|decreasing} Specifies the allocation method used for the signaling group. Possible values are:
• Cyclical – Trunks are allocated cyclically.
• Increasing – Trunk assignment according to increasing trunk numbers. The preference is for the allocation of the idle trunk with the lowest number.
• Decreasing – Trunk assignment according to decreasing trunk numbers. The preference is for the allocation of the idle trunk with the highest number.

3.1.5.1.5 Define CAS Configuration

Description

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This command can define cas trunk group configuration, such as trunk direction,allocation method, and others.

Syntax
define cas config <trunk-group-id> {bi-directional|incoming|outgoing} <cpc> {yes|no} {yes|no} {yes|no} {yes|no} <oos-threshold>

Arguments
trunk-group-id Specifies trunk group ID, ranging from 1 to 480 (ex: 1)

bi-directional|incoming|outgoing Specifies the supported direction for the calls.

cpc Specifies the calling party category to be sent on calls originated in corporate trunk groups. Possible values are II-1 to II-7 and "pass-through". Default is II-1. If provisioned as II-1 to II-7 the provisioned value will be sent in place of the received value in the R2 signaling. If provisioned as "passthrough" the value received in the R2 signaling will be sent unchanged.

yes|no Specifies if calling party number should be requested on TDM side or default number should be sent instead. Possible values are Yes or No. If set to Yes, the calling party number to be sent in the SIP side would be requested from the far-end equipment connected in the TDM side. If set to No, the calling party number to be sent in the SIP side would be the default number assigned to the corporate trunk. For trunks connected to class 5 switches it should be set to Yes. For trunks connected to R2 PBX it could be provisioned as Yes or No. The default value is Yes.

yes|no Specifies if called party condition B-5 should be replaced by B-1 on calls terminated to corporate trunk group. Possible values are Yes or No. If set to Yes, the called party condition B-5 is changed to B1 if received. If set to No, the called party condition B-5 is passed unchanged if received. For trunks connected to class 5 switches it should be set to No. For trunks connected to R2 PBX it could be provisioned as Yes or No. The default value is No.

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yes|no Specifies if the trunk-group is restricted for automatic collect call termination. Possible values are Yes or No. If set to Yes, ACC received in SIP side are not allowed to be completed. If set to No, ACC received in SIP side is allowed to be completed.For trunks connected to class 5 switches it should be set
to No. For trunks connected to R2 PBX it could be provisioned as Yes or No. The
default value is No.

yes|no Specifies if calls received from TDM side should be allowed to continue in
case the calling party number is not provided. Possible values are Yes or No. If
set to No, calls originated in the TDM side and for which the CgPN was not provided are allowed to continue. If set to Yes, calls originated in the TDM side
and for which the CgPN was not provided are not allowed to continue. For trunks
connected to class 5 switches it could be set to Yes or No. For trunks connected
to R2 PBX it should be provisioned as Yes. The default calling party number is
used on calls originated from R2 PBX for the cases where the CgPN was not provided. The default value is Yes.

oos-threshold Specifies the threshold to generate an alarm informing that the number of out of services trunks in the trunk-group has reached an alarming condition. The range is from 10 to 90 percent. The default value is 60 percent. Note: This command is only supported on the active FSB board.

3.1.5.1.6 create cas cpc-mapped

Description
Create a Calling Party Category (CPC) mapping to do the replacement during signaling exchange.

Syntax
create cas cpc-mapped <trunk-group-id> <1|2|3|4|5|6|7|8|11>
<1|2|3|4|5|6|7|8|11>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)

(cpc)<1|2|3|4|5|6|7|8|11>: Calling Party Category (corresponding to II-1..II-11) to be replaced by the third parameter.

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(cpc to be sent)<1|2|3|4|5|6|7|8|11>: Calling Party Category (corresponding to II-1..II-11) to be sent in the place of the second parameter.

Availability
This command is only supported on the active FSB board.

Example
create cas cpc-mapped 15 1 8

Note
Use view trunk group config <trkgrp-id> to view the CPC mapping configured.

3.1.5.1.7 remove cas cpc-mapped

Description
Remove a Calling Party Category (CPC) mapping.

Syntax
remove cas cpc-mapped <trunk-group-id> <1|2|3|4|5|6|7|8|11>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)

(cpc)<1|2|3|4|5|6|7|8|11>: Calling Party Category (corresponding to II-1..II-11) which mapping will be removed. It represents the origianl CPC, NOT the mapped value.

Availability
This command is only supported on the active FSB board.

Example
remove cas cpc-mapped 15 1

3.1.5.1.8 create cas cdc-mapped

Description
Map a called condition replacement during signal exchange.

Syntax
create cas cdc-mapped <trunk-group-id><input|output> <1|2|3|4|5|6|7|8> <1|2|3|4|5|6|7|8>

Description
Create a Called Party Condition mapping to do the replacement during signaling exchange.

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)

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<input|output>: direction of mapped-cdc related to this equipment where:
input when the cdc to be sent will be mapped (to-tele calls)
output when the cdc to be sent will be mapped (to-net calls)
(cdc)<1|2|3|4|5|6|7|8>: Called Party Condition (corresponding to B-1..B-8) to be replaced by the third parameter.

(cdc to be sent)<1|2|3|4|5|6|7|8>: Called Party Condition (corresponding to B-1..B-8) to be sent in the place of the second parameter.

Availability
This command is only supported on the active FSB board.

Example
create cas cdc-mapped 15 input 1 3

Note
Use view trunk group config <trkgrp-id> to view the Called Party Condition mapping configured.

3.1.5.1.9 remove cas cdc-mapped

Description
Remove a Called Party Condition mapping

Syntax
remove cas cdcondition-mapped <trunk-group-id><input|output> <1|2|3|4|5|6|7|8>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
<input|output>: direction of mapped-cdc related to this equipment to where:
input when the cdc received will be mapped (to-tele calls)
output when the cdc to be sent will be mapped (to-net calls)
(cdc)<1|2|3|4|5|6|7|8>: Called Condition (corresponding to B-1..B-8) which mapped will be removed. The second parameter represents the original called condition, NOT the mapped value.

Availability
This command is only supported on the active FSB board.

Example
remove cas cdc-mapped 15 output 1

3.1.5.1.10 define cas seizure ack timer

Description
Specifies timer value for Reception of SEIZURE ACK signal

Syntax

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define cas seizure ack timer <trunk-group-id> <timerval>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
<timerval> Specifies timer interval, in milliseconds in the range of 100-1000, multiple of 10. Default value is 200 miliseconds.

Availability
This command is only supported on the active FSB board.

Example
define cas seizure ack timer 1 200

3.1.5.1.11 define cas disconnection ack timer

Description
Specifies timer value for Reception of DISCONNECTION ACK signal

Syntax
define cas disconnection ack timer <trunk-group-id> <timerval>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
<timerval> Specifies timer interval, in seconds in the range of 0-180. Default value is 90 seconds.

Availability
This command is only supported on the active FSB board.

Example
define cas disconnection ack timer 1 60

3.1.5.1.12 define cas forced release ack timer

Description
Specifies timer value for Reception of FORCED RELEASE ACK signal

Syntax
define cas forced release ack timer <trunk-group-id> <timerval>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
<timerval> Specifies timer interval, in seconds in the range of 0-180. Default value is 90 seconds.

Availability
This command is only supported on the active FSB board.

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Example
define cas forced release ack timer 1 90

3.1.5.1.13 define cas presence forwarding signal timer

Description
Specifies timer value for Presence of Forwarding signal

Syntax
define cas presence forwarding signal timer <trunk-group-id> <timerval>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
<timerval> Specifies timer interval, in seconds in the range of 0-30. Default value is 15 seconds.

Availability
This command is only supported on the active FSB board.

Example
define cas presence forwarding signal timer 1 15

3.1.5.1.14 define cas reception forwarding signal timer

Description
Specifies timer value for Reception of Forwarding signal

Syntax
define cas reception forwarding signal timer <trunk-group-id><timerval>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
<timerval> Specifies timer interval, in seconds in the range of 0-20. Default value is 7 seconds.

Availability
This command is only supported on the active FSB board.

Example
define cas reception forwarding signal timer 1 7

3.1.5.1.15 define cas answer timer

Description
Specifies timer value for Answer on terminated calls.

Syntax
define cas answer timer <trunk-group-id> <timerval>

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Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
<timerval> Specifies timer interval, in seconds in the range of 0-180. Default value is 90 seconds

Availability
This command is only supported on the active FSB board.

Example
define cas answer timer 1 90

3.1.5.1.16 define cas reanswer timer

Description
Specifies timer value for Re-answer on terminated calls

Syntax
define cas reanswer timer <trunk-group-id><timerval>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)
<timerval> Specifies timer interval, in seconds in the range of 0-180. Default value is 90 seconds

Availability
This command is only supported on the active FSB board.

Example
define cas reanswer timer 1 90

3.1.5.1.17 define cas trunk status

Description
Set the state of a trunk range to either blocked, unblocked or immediate

Syntax
define cas trunk status <first_trunk> <last_trunk> <blocked | unblocked | immediate>

Arguments
<first_trunk> First trunk of range. Format {e1|t1}.scm.<port>.<channel>
<port> E1 identification, values from 1 to 16.
<channel> Time slot identification, values from 1 to 31. Value 16 is not valid since it is used for signaling.

<last_trunk> Last trunk of range. Format {e1|t1}.scm.<port>.<channel>

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<port> E1 identification, values from 1 to 16.
<channel> Time slot identification, values from 1 to 31. Value 16 is not valid since it is used for signaling.

<blocked|unblocked> Specifies the status of R2 trunks

Availability
This command is only supported on the active FSB board.

Example
define cas trunk status e1.scm.1.1 e1.scm.1.31 unblocked

3.1.5.1.18 define cas country variant <country>

Description
Set the country variant for CAS

Syntax
define cas country variant <country>

Arguments
<country> Specifies country id. (ex: brazil)

Availability
This command is only supported on the active FSB board.

Example
define cas country variant brazil

3.1.5.1.19 view cas trunk status

Description
View the current status of following trunk range

Syntax
view cas trunk status *<first_trunk> <last_trunk>*

Arguments
<first_trunk> First trunk of range. Format {e1|t1}.scm.<port>.<channel>
<port> E1 identification, values from 1 to 16.
<channel> Time slot identification, values from 1 to 31. Value 16 is not valid since it is used for signaling.

<last_trunk> Last trunk of range. Format {e1|t1}.scm.<port>.<channel>
<port> E1 identification, values from 1 to 16.
<channel> Time slot identification, values from 1 to 31. Value 16 is not valid since it is used for signaling.

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Availability
This command is only supported on the active FSB board.

Example
view cas trunk status e1.scm.1.1 e1.scm.1.31

3.1.5.1.20 view cas statistics

Description
Shows cas statistics

Syntax
view cas statistics

Arguments
None

Availability
This command is supported on both active and standby FSB board.

Example
view cas statistics

CAS Statistics

ATTEMPTS_TO_ALLOCATE_MF 10
ALLOCATED_MF 07
AVERAGE_WAITING _TIME_MF 10
AVERAGE_RETENTION_TIME_MF 13

clear cas statistics

Syntax
clear cas statistics

Description
Reset the current CAS statistics

Arguments
None

Availability
This command is supported on both active and standby FSB board.

Example
clear cas statistics

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3.1.5.1.21 view cas trunk group statistics

Description
View CAS trunk group statistics

Syntax
view cas trunk group statistics <trunk-group-id>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)

Availability
This command is supported on both active and standby FSB board.

Example
view cas trunk group statistics 1

CAS Trunk Group Statistics

ATTEMPTS_TO_ALLOCATE_R2_TRUNK 10
ALLOCATED_R2_TRUNK 07
R2_TRUNK_AVERAGE_TRAFFIC 10
R2_TRUNK_AVERAGE_RETENTION_TIME 13
UNSUCESSFUL_CALLS_UNEXPECTED_SIGNAL 10
UNSUCESSFUL_CALLS_EXPIRY_PRESENCE_FW 20
UNSUCESSFUL_CALLS_EXPIRY_RECEPTION_FW 20
UNSUCESSFUL_CALLS_FORCED_RELEASE 24

3.1.5.1.22 clear cas trunk group statistics

Description
Reset the current CAS trunk group statistics

Syntax
clear cas trunk group statistics <trunk-group-id>

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1)

Availability
This command is supported on both active and standby FSB board.

Example
clear cas trunk group statistics 1

3.1.5.1.23 view trunk group config

Description

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View the current CAS trunk group configuration

Syntax
view trunk group config [trunk-group-id]

Arguments
<trunk-group-id> Specifies trunk group ID, ranging from 1 to 480 (ex: 1). This is an optional parameter. If no trunk group provided, will show a summary of all configured trunk groups.

Availability
This command is supported on both active and standby FSB board.

Example
view trunk group config 8

Trunk Group 8 :

Trunks:
e1.scm.1.1 to e1.scm.1.31
e1.scm.9.1 to e1.scm.9.31

Signaling: R2
Allocation Method: Cyclical
Trunk Direction: Bi-directional
CPC Corporate: II-1
Request CGPN: yes
Called Party Condition Replacement: no
Acc Restriction: no
Release No CGPN: yes
OOS Threshold: 60
Cpc 11-TransferredCall mapped to 01-RegularSubscriber
Called condition B4-Congestion mapped to B7-Vacant number
Called condition B5-Free without charging mapped to B1-Free with charging

********** Timers **********

Seizure Ack: 200 ms
Disconnection Ack: 90 seconds
Forced Release Ack: 90 seconds
Presence Forwarding Signal: 15 seconds
Reception Forwarding Signal: 7 seconds
Answer: 90 seconds
Reanswer: 90 seconds

3.1.5.1.24 define cas cpc

Description
Specifies CAS cpc

Syntax
define cas cpc <II-1> <II-2> <II-3> <II-4> <II-5> <II-6> <II-7>

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Arguments
<II-1> Specifies the syntax used in SIP for the category defined by the R2 signal II-1. The default value is ordinary.

<II-2> Specifies the syntax used in SIP for the category defined by the R2 signal II-2. The default value is specchg.

<II-3> Specifies the syntax used in SIP for the category defined by the R2 signal II-3. The default value is test.

<II-4> Specifies the syntax used in SIP for the category defined by the R2 signal II-4. The default value is payphone.

<II-5> Specifies the syntax used in SIP for the category defined by the R2 signal II-5. The default value is operator.

<II-6> Specifies the syntax used in SIP for the category defined by the R2 signal II-6. The default value is data.

<II-7> Specifies the syntax used in SIP for the category defined by the R2 signal II-7. The default value is payphoneIU.

Availability
This command is only supported on the active FSB board

Example
define cas cpc ordinary specchg test payphone operator data payphoneIU

3.1.5.1.25 view cas config

Description
View the current CAS configuration

Syntax
view cas config

Arguments
None

Availability
This command is supported on both active and standby FSB board.

Example
view cas config

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Configuration for CAS

Country Variant: Brazil
CPC: ordinary specchg test payphone operator data payphoneIU

3.1.5.1.26 define cas trunk monitoring

Description
Define cas trunks for monitoring.

Syntax
define cas trunk monitoring {all|trunk-group|range}[trunk-group-id-or-first-trunk][last-trunk]

Arguments
{all|trunk-group|range}:
all: all provisioned cas trunks will be monitored.
trunk-group: all provisioned trunks of the trunk group id will be monitored since it's cas provisioned.
range: all provisioned cas trunks within the trunk range will be monitored.
[trunk-group-id-or-first-trunk]:
the trunk group id since <all|trunk-group|range> is trunk-group
the first trunk of the range since <all|trunk-group|range> is range
(In case of channel equals 0 (ex. e1.scm.9.0) the whole link is assumed)
[last-trunk]:
the last trunk of the range since <all|trunk-group|range> is range
(case this parameter was ommited, last-trunk is assumed equals first-trunk)
(In case of channel equals 0 (ex. e1.scm.9.0) the whole link is assumed)

Availability
This command is supported on both active and standby FSB board.

Example
define cas trunk monitoring all
Trunk e1.scm.2.1 monitoring defined
Trunk e1.scm.2.2 monitoring defined
Trunk e1.scm.2.2 monitoring defined
…

define cas trunk monitoring trunk-group 18
Trunk e1.scm.3.3 monitoring defined
Trunk e1.scm.3.4 monitoring defined
Trunk e1.scm.3.5 monitoring defined

define cas trunk monitoring range e1.scm.3.3
Trunk e1.scm.3.3 monitoring defined

define cas trunk monitoring range e1.scm.3.3 e1.scm.3.5
Trunk e1.scm.3.3 monitoring defined
Trunk e1.scm.3.4 monitoring defined

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Trunk e1.scm.3.5 monitoring defined

define cas trunk monitoring range e1.scm.3.0
Trunk e1.scm.3.1 monitoring defined
Trunk e1.scm.3.2 monitoring defined
…
Trunk e1.scm.3.31 monitoring defined

3.1.5.1.27 remove cas trunk monitoring

Description
Remove monitoring of cas trunks.

Syntax
remove cas trunk monitoring {all|trunk-group|range}[trunk-group-id-or-first-trunk][last-trunk]

Arguments
{all|trunk-group|range}:
all: all provisioned cas trunks will not be monitored.
trunk-group: all provisioned trunks of the trunk group id will not be monitored since it's cas provisioned.
range: all provisioned cas trunks within the trunk range will not be monitored.
[trunk-group-id-or-first-trunk]:
the trunk group id since <all|trunk-group|range> is trunk-group
the first trunk of the range since <all|trunk-group|range> is range
(In case of channel equals 0 (ex. e1.scm.9.0) the whole link is assumed)
[last-trunk]:
the last trunk of the range since <all|trunk-group|range> is range
(case this parameter was ommited, last-trunk is assumed equals first-trunk)
(In case of channel equals 0 (ex. e1.scm.9.0) the whole link is assumed)

Availability
This command is supported on both active and standby FSB board.

Example
remove cas trunk monitoring all
Trunk e1.scm.2.1 monitoring removed
Trunk e1.scm.2.2 monitoring removed
Trunk e1.scm.2.2 monitoring removed
…

define cas trunk monitoring trunk-group 18
Trunk e1.scm.3.3 monitoring removed
Trunk e1.scm.3.4 monitoring removed
Trunk e1.scm.3.5 monitoring removed

remove cas trunk monitoring range e1.scm.3.3
Trunk e1.scm.3.3 monitoring removed

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remove cas trunk monitoring range e1.scm.3.3 e1.scm.3.5
Trunk e1.scm.3.3 monitoring removec
Trunk e1.scm.3.4 monitoring removed
Trunk e1.scm.3.5 monitoring removed

remove cas trunk monitoring range e1.scm.3.0
Trunk e1.scm.3.1 monitoring removed
Trunk e1.scm.3.2 monitoring removed
…
Trunk e1.scm.3.31 monitoring removed

3.1.5.1.28 view cas trunk monitoring

Description
View cas trunks being monitored or monitored data collected.

Syntax
view cas trunk monitoring [all|trunk-group|range][trunk-group-id-or-first-trunk][last-trunk]

Arguments
[all|trunk-group|range]:
if ommited: show a table of trunks with monitored provisioning summary:
=> indicates that the trunk is not being monitored
m => indicates that the trunk is being monitored but has no data collected yet
M => indicates that the trunk is being monitored and also has data collected
Otherwise, show data collected for:
all: all trunks.
trunk-group: all trunks of the trunk group id.
range: all trunks within the trunk range.
[trunk-group-id-or-first-trunk]:
the trunk group id case <trunk-group>
the first trunk case <range>
(channel equals 0 (ex. e1.scm.9.0) the whole link is assumed)
[last-trunk]: the last trunk case <range>
(case this parameter was ommited, last-trunk is assumed equals first-trunk)

Availability
This command is supported on both active and standby FSB board.

Example
view cas trunk monitoring

```
Link Addr     Grp   Sign   Channel Table
------------  ---  ------  12345678901234567890123456789012
e1.scm.1.0                 ................................
e1.scm.2.0     15      r2  .MMMMMMMMMMMMMMMM.MMMM..........
e1.scm.3.0                 ................................
```

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```
e1.scm.4.0              .....
e1.scm.5.0              .....
e1.scm.6.0              .....
e1.scm.7.0              .....
e1.scm.8.0              .....
e1.scm.9.0              .....
e1.scm.10.0             .....
e1.scm.11.0             .....
e1.scm.12.0             .....
e1.scm.13.0             .....
e1.scm.14.0             .....
e1.scm.15.0             .....
e1.scm.16.0             .....

view cas trunk monitoring all
Link Addr     Grp  Calls
------------  ---  -----
e1.scm.2.2    15     2

   #  Start                  Duration
 ---  -----------------  --------------------------
   1  29/Jun/09 17:15:46                 0:00:00
            Calling: 88884567
            Called: 35011031


   ==> DISCONNECTION                    0:00:00
   <== DISCONNECTION ACK                   0:00:00

   #  Start                  Duration
 ---  -----------------  --------------------------
   2  29/Jun/09 17:15:46  (in progress)      0:00:00

   ==> SEIZURE                         0:00:00
   <== SEIZURE ACK                        0:00:00
   ==> DISCONNECTION                      0:00:00

   ==>   I-9                          0:00:00
   <==   A-1                           0:00:00
   <==   A-4                           0:00:00
…
```

Other examples (results ommited):
view cas trunk monitoring trunk-group 18
view cas trunk monitoring range e1.scm.2.2
view cas trunk monitoring range e1.scm.3.3 e1.scm.3.5
view cas trunk monitoring range e1.scm.3.0

3.1.5.1.29 define cas acc-to-cpc

Description:
Set whether Automatic Collect Call should be mapped to Calling Party Category

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Syntax:
define cas acc-to-cpc <trunk-group-id> <yes|no>
Arguments:
<trunk-group-id> range: 1-480
(should map?)<yes|no>: answer to question: Should acc be mapped to cpc?

Availability:
This command is supported on both active and standby FSB board.

Example:
A7515S31_57:ACT-FSB:1.1(r0)>=8:diag:main# define cas acc-to-cpc 1 yes
This command is not allowed for ISDN trunk groups
A7515S31_57:ACT-FSB:1.1(r0)>=9:diag:main# define cas acc-to-cpc 3 yes
Mapping of Automatic Collect Call for trunk group 3 defined to yes

3.1.5.1.30 define cas div-to-cpc

Description:
Set whether Diversion header and History-info should be mapped to Calling Party Category.

Syntax:
define cas div-to-cpc <trunk-group-id> <yes|no>

Arguments:
<trunk-group-id> range: 1-480
(should map?)<yes|no>: answer to question:Should Diversion header and History-info be mapped to cpc?

Availability:
This command is supported on both active and standby FSB board.

Example:
A7515S31_57:ACT-FSB:1.1(r0)>=10:diag:main# define cas div-to-cpc 1 yes
This command is not allowed for ISDN trunk groups
A7515S31_57:ACT-FSB:1.1(r0)>=11:diag:main# define cas div-to-cpc 3 yes
Mapping of Diversion header and History-info for trunk group 3 defined to yes

FEATURES RELEASED IN Hardening

As one of the key components of Alcatel's Next Generation Network (NGN) solutions, the Alcatel 7515S Media Gateway (MG) provides seamless inter-working of voice connections between the public switched telephone network (PSTN) and IP networks. Voice over IP solutions enable service providers to address the challenges they face today, and evolve their infrastructure

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towards NGNs. Alcatel understands that the successful deployment of NGNs depends on the seamless mediation of services between PSTN and IP networks.

Designed for deployment in low-density VoIP trunking Points of Presence (PoPs), the A75153.1S is a carrier grade media gateway locates between PSTN and IP network. Making use of open standardized interfaces, the Alcatel 7515 MG can be integrated into any multi-vendor next-generation network.

It is noted A7515 SIP Gateway has two releases (R2.0S and R3.0S) at the time being.
SIP Signaling and Control. (Only new functionalities listed here, comparison with R2.0S.)

- Supported methods: Update, Refer.

Supplementary services (only new functionalities listed here, comparison with R2.0S.)

- Call Transfer.

3.1.6 T.38 Fax

- Upon detection of the fax transmission(CNG/V21 event), the receiving gateway initiates a switch to T.38 fax mode by sending a re-INVITE request with T.38 SDP
- The gateway switches to T.38 fax mode after receiving a re-INVITE request with T.38 SDP
- The session switches back to audio mode at the end of the fax transmission

3.1.7 Call Transfer Supplementary Service

- The Call Transfer supplementary service enables a user to transfer an established call to a third party. For the original call, the “served user” may have been either the calling or called party. This service differs from the Call Forwarding supplementary services in that the latter deals only with incoming calls which have not yet reached the “fully-established” state, whereas in the case of Call Transfer an established end-to-end connection exists. This release supports four types of call transfer:
    - Basic Transfer
    - Exposing Target transfer
    - Protecting Target transfer
    - Attended Transfer
- 7515 R3.1S Gateway can be a Transferee or Transfer target but not a Transferor(originator)

7515 R3.0S Gateway is compliant to draft-ietf-sipping-cc-transfer-03.

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3.1.8 Digit manipulation on calling number

A7515 R3.0S GW supports that, for incoming SIP calls to PSTN network, the calling party number can be manipulated according to configured DOM dial plans. It is noted the called party number can be manipulated too which is already supported by A7515 R2.0S GW.

- Support up to 128 total number translation rules
- Match the calling number according to prefix
- Remove the prefix digits from the calling number according to the rules
- Insert predefined digits in the begin of the calling number

3.1.9 Diversion header

- Compliant to draft-levy-sip-diversion-05.txt
- When multiple diversion headers are presented, the top-most Diversion header is used to set the bottom-most ISDN Redirecting number information element. The bottom-most Diversion header is used to set the top-most ISDN Redirecting number information element.
- The top-most ISDN Redirecting number information element is used to set the bottom-most SIP Diversion header. The bottom-most ISDN Redirecting number information element is used to set the top-most SIP Diversion header.
- Presence of the diversion-counter in each of the Diversion headers is optional. And, there is no Redirection counter associated with this ISDN information element. From SIP to PRI direction, neglect it. From PRI to SIP direction, always set diversion-counter to 1

3.1.10 Network Failure

- Disconnect VoIP calls if no Real-Time Control Protocol (RTCP) packets are received within a configurable time period.
- Remote side must support RTCP if network failure is required.

3.1.11 Control and Data use the same IP port

A7515 R3.1S GW support receiving/sending SIP signalling messages with the same IP port of sending/receiving RTP traffic. This feature is not supported in R2.0S.

3.1.12 UDP interception on NPU

This is requested by TEC/TAC according to the practice in fields.

In order to narrow down the investigating area, TEC/TAC need to capture the RTP stream received/sent on the IP interface. This requirement will be

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implemented mainly on NPU.

User case

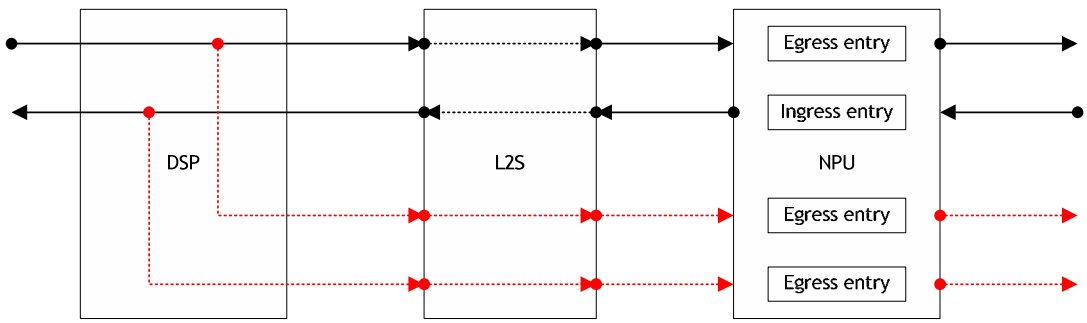
The use case is:

- User provisions a notebook on a IP network with the access to the 7515 data port at IP layer
- User launches a Ethereal like tools to capture the packets intercepted from 7515
- User configures an UDP interception descriptor. The descriptor includes following attributes:
  —Target IP address: to which the intercepted packet will be cast. IP address used by system itself shall be rejected
  —Target base UDP port:
  — State: idle, busy, disabled
- User issues a command to 7515 to start the interception. The command may be
  #diag intercept udp context <context ID>
  #diag intercept udp next context
- System will duplicate the UDP packet received/sent by NPU at line side to the address specified by the interception descriptor
- User issues command
  #diag intercept udp stop
  to stop the interception

3.1.13 RTP interception on DSP (DSP diagnostic)

This is to enable a MSPD Miro build-in diagnostic function. MSPD should adapt this feature to our VCID environment so that host can drive Miro correctly.

This function is simply shown in the diagram below:



User case

The use case is similar with UDP interception:

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- User provisions a notebook on a IP network with the access to the 7515 data port at IP layer
- User launches a Ethereal like tools to capture the packets intercepted from 7515
- User configures an RTP interception descriptor. The descriptor includes following attributes:
  —Target IP address: to which the intercepted packet will be cast. IP address used by system itself shall be rejected
  — Target base UDP port:
  —Source IP address
  —Source base UDP port
  — State: idle, busy, disabled
- User issues a command to 7515 to start the interception. The command may be
  #diag intercept rtp context <context ID>
  #diag intercept rtp next context
- system will duplicate the rtp packet received/sent by DSP at its packet side to the address specified by the rtp interception descriptor
- User issues command
  #diag intercept rtp stop
  to stop the interception

3.1.14 NPU statistics per flow

This is required by TEC and is helpful to trouble-shooting.

The NPU is expected to provide following statistics per VCID in both direction separately.

Apart from statistics per flow, overall statistics shall be provided too like below

Direction	Item	Description	Remark
Egress	AllPaReE	Number of packets received from backplane side	7515 NPU provide statistics according to RFC 2819 (RMON) and RFC 2665 (MIB) which cover the requirement listed here
	AllPaSeE	Number of packets sent to line side	
Ingress	AllPaReI	Number of packets received from line side	

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	AllPaSeI	Number of packets sent to backplane side	

All these statistics items can be reset respectively on request of host.

3.1.15 Alarm history

- Provide alarm history.
- The display of alarm history may be filtered by specifying matching string.
-

3.1.16 T1 alarm

Implementation on E1/T1 specific alarm is shown below:

Alarm name	E1 HDS	T1 HDS
Loss Of Signal (LOS)	Supported	Supported
Loss Of Frame (LOF)	Supported	Supported
Remote Alarm Indication (RAI)	Supported	Supported
Alarm Indication Signal (AIS)	Supported	Supported
Slip	Supported	Supported
Line code violations	Supported	TBI
Frame bit errors	TBI	TBI
CRC-4 bit errors	TBI	TBI

TBI: To Be Implemented

3.1.17 Auto speech path tracing

The function is to have the system to monitor the quality of the speech path and take the traces along the path when detecting quality issues.

The quality relevant issues may include:

- One-way audible voice
- Intermittent voice
- Noise

3.1.18 Auto DSP trace

Make system take DSP trace automatically when predefined condition is met.

The predefined condition may be:

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- No response on the heartbeat message from DSP to host
- Specific alert event is received from DSP

The traces include 4 files to be stored on CF card:

- Iram.dmp (?KB)
- Eram.dmp (~1MB)
- SDram.dmp (~16MB)
- History.trc (~KB)

Where there is no enough room on CF card for storing the trace, the execution shall be aborted.

3.1.19 IP Filter （Firewall）

IP Filter feature in A7515 R3.1S is to modify the default action to drop packet when a packet arrives but cannot match any filter rule.

In the new implementation, if IP filtering is enabled but packet is not matched to any filter, packet will be dropped. We recommend configure the IP filter rules traffic by traffic and reduce the range of source IP addresses to minimize the security risk.

3.1.20 Security Threats for Network including DOS

A7515 MGW should have the basic ability to prevent itself from Signalling/Data DOS attack according to Alcatel Security Policy.
In A7515 R3.1S, only flooding protection for Protocol will be implemented, when flood attack, this mechanism reduces the system performance. To reduce the CPU occupancy when protecting MGW from flooding attack

3.2 Features released in 2.0S:

3.2.1 TDM network interfaces

- 8/16 E1 interface
- or 8/16 T1 interface

3.2.2 Packet network interfaces

- 100 Mb/s Ethernet for VoIP
- 10/100 Mb/s Ethernet for control, signaling and element management

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3.2.3 PSTN signaling interface

- PRI signaling (ETSI/ANSI)

protocol	Meaning	User side / Network side
PRI_4ESS	AT&T	ATT User side + Network side
PRI_5ESS	AT&T	
PRI_DMS100	Northern Telecom	NTI User side + Network side
PRI_KDDNTT	Japanese	NTI User side
PRI_NET5	ETSI(Telenetworks:Belgian)	NET5 User side +Network side
PRI_NET5SW	ETSI Swissnet	
PRI_Q.SIG	ETSI QSIG	

- Configurable User Termination side or Network Termination side
- Configurable dial plan on different E1/T1 links.

3.2.4 SIP Signaling and Control

- Compliant to RFC 3261/3262/3264 & SDP RFC 2327
- Supported methods: INVITE, CANCEL, BYE, ACK, REGISTER, INFO.
- Supported responses: 1xx/2xx/3xx/4xx/5xx/6xx.
- Configurable SIP transaction timers.
- Works with Proxy or without Proxy. If the primary Proxy fails, automatically switches to the secondary one.
- Modifying connection parameters for an already established call (re-INVITE).
- Works with a Redirect server and handling 3xx responses.
- Early Media.
- PRACK reliable provisional responses (RFC 3262).
- Session Timer (draft-ietf-sip-session-timer-10.txt).
- DTMF out of band transfer using INFO method
- Reliable UDP transport, with retransmissions.
- Negotiates codec from a list of given codec list.

3.2.5 PSTN-to-SIP Inter-working

- Definition of trunk group for routing incoming SIP calls to PSTN network.
- Inter-working of PSTN to SIP Cause Code Mapping
- PSTN to SIP called number can be manipulated according to dial plans.
- SIP to PSTN Called number can be manipulated according to dial plans.
- Provides local-gateway hairpinning.

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3.2.6 Basic VoIP/ FoIP Services

- PSTN phone to PSTN phone
- Fax services
- SIP Residential gateway/SIP software phone to PSTN phone
- PSTN phone to Residential gateway/ SIP software phone.
- Configurable call policy on different E1/T1 in-link/out-links
- Configurable network profile on different call policies
- Configurable VoIP dial plan on different call policies.

3.2.7 Supplementary services

- Call hold
- Call Forwarding

3.2.8 Voice Codecs

- G.711 (64 kb/s, µ-law and A-law, with/without Silence suppression)
- G.723.1 (6.3 kb/s)
- G.729A (8 kb/s)
- Automatic Voice/Fax/Data codec switchover

3.2.9 Fax

- Automatic FAX detection and voice/fax switchover
- Fax bypass, G.711 transparent pass-through

3.2.10 Voice processing

- Echo cancellation, G.168 compliant with up 128 ms tail length
- Configurable jitter buffer, with configurable upper limit of up to 200 ms.
- Configurable voice frame size, from 5 to 30ms with 5ms steps
- Configurable silence suppression
- Comfort noise generation
- DTMF transport
  - ✓ DTMF relay inband via media stream (RTP).
  - ✓ DTMF relay outband via RFC2833
  - ✓ DTMF relay via Signaling message: INFO
  - ✓ DTMF relay via both media stream(RTP) and Signaling message: INFO
  - ✓ DTMF relay via both media RFC2833 and Signaling message: INFO
- Call progress tone generation

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3.2.11 Routing

- RIP, OSPF, OSPFv2
- IGMP
- Static routes
- IP route and packet filtering
- Domain Name Service
- Network Time Protocol V4

3.2.12 Quality of Service

- IP ToS

3.2.13 System and Network Management

- Telnet
- RS-232 console
- User/Password with multiple access rights
- User command line interface
- TFTP remote file transfer for software upgrade
- SNMP v2 and SIP MIB supporting
- Visual and audible Alarms
- SIP protocol message statistics
- Call statistics per connection
- Real time call trace via SYSLOG
- CPU/Memory usage

3.2.14 Capacity

- 240/480(E1 configuration) or 192/384(T1 configuration) VoIP/FoIP connections simultaneously
- 4CAPS
- G711:480 calls (2 or 4 DSP chips in DSP card)
- G723:320 calls (4 DSP chips in DSP card)
- G729:400 calls (4 DSP chips in DSP card)

3.2.15 Non-hitless Redundancy

- Boot sequence control of two FSBs
- RTM boot controlled by the active FSB
- Sync configuration file between two FSBs
- Inter-communication between two FSBs
- Monitor control/data Ethernet ports;
- Monitor software/hardware status, switchover in case of failure.
- After switchover, recover communication with Switch/Hub/Router

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- After switchover, recover SIP stack and communication with its registrars/proxies.
- After switchover, recover ISDN stack and communication with PSTN network.

4 UI Command

Refer to [3FZ045565002ACZZA] 7515R3.1S User Manual

5 Bug-fix History

None

6 Installation and Upgrade Procedures

6.1 Installation

Refer to [3FZ045565005ACZZA] 7515R3.1S Quick Start

6.2 *Installation and Upgrade Procedures*

6.2.1 Upgrade Procedures for upgrading from R2.0S to R3.1S

1. Prepare a PC as a TFTP server, and TFTP software

Make sure 7515 can download system from PC with TFTP software running.

2. Connect the PC to active FSB board via CTRAL port (baud rate is 115200)

The following command can help to know which FSB board is active.

```
My-Chassis:ACT-FSB:1.1(r0)>=10:diag:main# vi ch
-------------------------------------------------------------
Ch VB-Sl    VB-Type  State    Function     LBI   VM-Type
-------------------------------------------------------------
1  2        VBSCM    UP       STANDBY-UP         VMSCM
1  1        VBSCM    UP       ACTIVE-UP   * scm  VMSCM
-------------------------------------------------------------
```

3. Backup system files before downloading new files

Generally, “config.txt “ and “dspfw_0.axf” should be backup on two FSB boards at first. Corresponding command as follow:

“rename <current-name> <new-name>”

For example:

“My-Chassis:ACT-FSB:1.1(r0)>=13:diag:main# rename config.txt config.old”

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Then insert “RC 1 2 “ to switch to other FSB board, do the same operation as done on board 1.1
(Insert “exit “ to exit from the standby board to active board)

4. Download “*.pak” 、”config.txt” and “dspfw_0.axf” files to 7515

Corresponding command as follow:
“ tftp get <ip-addr:rmt-file> [[ff:]<lcl-file>]”
On the active FSB board, download system files from TFTP server.
For example:
“ My-Chassis:ACT-FSB:1.1(r0)>=16:diag:main# tftp get 10.10.0.61:config.txt”

On the standby FSB board , only can download system files from the active board . 7515 assigns different internal ip address to two board , “200.1.1.33” for board 1.1 ,and “200.1.1.34” for board 1.2. So ,if download files from board 1.1 to board 1.2 ,you can issue:
“My-Chassis:rem-cons:STB-FSB:1.2(r0)>=2:sduc:main# tftp get 200.1.1.33:config.txt “

5. Define the nvram files on two boards

UI command:

Main# define nvram pf <filename> [<filename>]

Access to standby FSB, use internal IP to get config.txt from active FSB, (Internal IP 200.1.1.33 represents downside FSB and 200.1.1.34 represents upside one). Redefine nvram configuration on the standby FSB board, and make sure the two boards use the same *.pak files.Insert “ view nvram “ to check whether filename on two boards is right.

Then directly reboot system without “save”. After the rebooting complete, insert “ view version “ on both FSB and check if the version is right.

6.2.2 Upgrade/Downgrade Notes for pak before and after (including S31B0106) of R3.1S

In case upgrade from the pak before S31B0106 to the pak S31B0106 or newer then S31B0106
(for example upgrade from S31B0105 to S31B0106 or S31B0107)
you need take following action:

1) Before upgrade, backup the config.txt. The backup config.txt shall be used once need to be downgrade from the pak S31B0106 or newer then S31B0106 to the pak before S31B0106
2) Use UI command to view all policy
3) Record the trunk group ID of each policy
4) Reboot with the new pak, redefine the call policy and save

in case the pak above S31B0106(including S31B0106) need to be upgraded, no additional action need to be done.

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7 Hardware

The Alcatel 75153.1S is a low-density media gateway that up to four chassis can be cascaded to support a larger “system”, delivering high availability and carrier grade quality transmission with redundant service processing modules, power supplies, fans, external synchronous clocks and packet interfaces. The Alcatel 75153.1S can offer the flexible interfaces to the TDM/packet network and the independence of the key modules facilitates the upgrade of the Alcatel 7515 to protect the customer’s investment.

The Alcatel 7515 Media Gateway embraces a modular architecture, which uses an open, programmable call control model. Call processing is independent of other functions, such as resource management, routing, or forwarding.

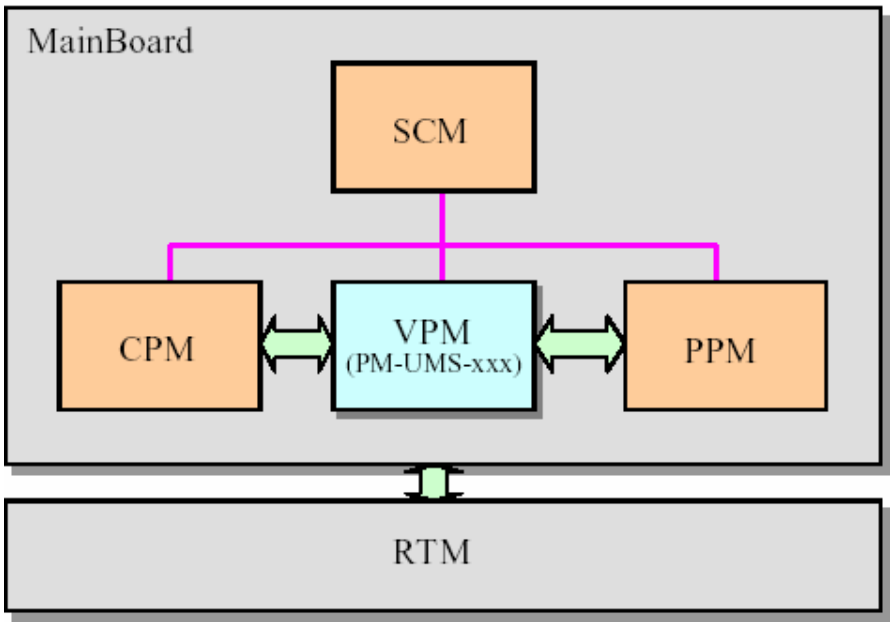

Figure 1 A7515 Non-Redundancy System Architecture Overview

The Alcatel 75153.1S integrates the basic common modules such as the circuit processing module, packet processing module and system control module into a single main-board, but separates the key modules - the voice processing module and rear transition module – from the main board to facilitate the upgrade, shown as figure 1.

Chassis Characteristics

- Height: 8.8 cm (3.48 inch)
- Width: 45 cm (17.7 inch, without brackets)

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- Depth: 46 cm (18.1 inch)
- Weight (shelf only): 10.5 kg
- Weight (fully configured): 13.5kg

Operating Environment
- Operating temperature range: 0 C to 40 C (32 F to 104 F)
- Storage temperature range: -20 C to 65 C (-4 F to 149 F)
- Operating humidity: 0% to 80%, non-condensing

Power
- Maximum 200W, 48V, DC power
- 1+1 redundant power supplies.
- Dual Power supply

Individual ON/OFF switch, power card and cooling fan

7.1 HW Requirements

The package is released with the following HW configurations (backward compatible hardware as supported also by software Rel2.3)

Description	Mnemonic	Code
Forwarding System Processl Module - C5 Processor Upgrade version of 3FZ30062AAAA.	FM-SPM-C5	3FZ30062AAAB (3FZ30062AAAA)
PSTN Interface w/ 8 E1/T1 and Packet Interface w/2 10/100 mbps routed Fast Ethernet ports	FM-WIM-8E1/2FE	3FZ30063AAAB (3FZ30063AAAA)
Universal DSP Module - 240 channels G.711	PM-UMS-240	3FZ30064ABAA
Universal DSP Module - 480 channels G.711	PM-UMS-480	3FZ30064AAAA
Universal DSP Module - 480 channels G.711, 400 channels G.729	PM-UMS-960	3FZ30064ADAA
The blank panel for FM-SPM-C5/FM-WIM-8E1/2FE board.When a slot is not filled by a board, it should be covered by this panel to provent the dust.	CE-Blank-Panel	3FZ40093AAAA
Front-Left Fan Tray	CE-FAN-Front-Left	3FZ40084AAAC
Front-Right Fan Tray	CE-FAN-Front-Right	3FZ40085AAAC
Rear Fran Tray	CE-FAN-Rear	3FZ40086AAAC
DC Power Tray, 48V, 10A	CE-PSU-DC	3FZ40087AAAC
2U A7515 Shelf - Full configuration (including 1 chassis(3FZ50047ABAC), 3 Fan Tray(3FZ40084,85,86AAAC) & 1 PowerTray(3FZ40087AAAC))	CE-SHF-7515-FL	3FZ50048AAAC

Table 7-1 Supported Hardware

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Optional Elements: Cables	Mnemonic	Code
Cable to connect to E1/T1 port of the FM-WIM-8E1/2FE, 3 Metre. Interface: RJ45 to RJ45(need work with Impedence Converter)	CE-CAB-E1/T1-3	3FZ45012ADAA
Cable to connect to 10/100Mbps Fast Ethernet port of the FM-WIM-8E1/2FE, 6Metre	CE-CAB-FE	3FZ45002AGAA
Cable to connect to Craft Serial port(RJ45) of the FM-WIM-8E1/2FE. Interface: RJ45 to RS232(female)	CE-CAB-CS	3FZ45003ADAA
Cable to connect to external clock input interface of the FM-WIM-8E1/2FE(1.544Mbps),0.5 Metre	CE-CAB-BITS	3FZ45004AAFA
Cable to connect to external clock input interface of the FM-WIM-8E1/2FE(2.048MHz),0.5 Metre	CE-CAB-SSU	3FZ45005AAFA
Power Cable(3 Metres)	CE-CAB-PW	3FZ45009ADAA

Table 7-2 optional elements: cables

8 Media Gateway Element Manager and Alarm Manager (Mandatory for SNMP feature, optional for others)

New MIB "pizzabox-r2.mib" defined for R2 traps.
The MIB “pizzabox-r2.mib” based on MIB “pizzabox-smi.mib”, so when want to compile it, should load MIB “pizzabox-smi.mib” firstly.

MIB defined as following:

```
PizzaBox-R2-MIB  DEFINITIONS ::=  BEGIN

IMPORTS
   MODULE-IDENTITY, OBJECT-TYPE, NOTIFICATION-TYPE,
   Integer32, TimeTicks, Counter32, Gauge32, IpAddress
      FROM SNMPv2-SMI
   TEXTUAL-CONVENTION, DisplayString, RowStatus
   FROM SNMPv2-TC
   pbv1 FROM PizzaBox-SMI;

r2  MODULE-IDENTITY
      LAST-UPDATED    "0810150000Z"
      ORGANIZATION    "ASB NGVN BU"
      CONTACT-INFO
```

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```
            "Phone number: +86-21-50554550;
             http://www.alcatel-sbell.com.cn"
        DESCRIPTION
            ""
        ::= { pbv1 11 }

r2Traps         OBJECT IDENTIFIER ::= { r2 0 }

r2TrapsObjects  OBJECT IDENTIFIER ::= { r2Traps 1 }

    r2AlarmTrunkGroupId OBJECT-TYPE
        SYNTAX      INTEGER
        MAX-ACCESS  not-accessible
        STATUS      current
        DESCRIPTION
            "r2 alarm trunk group Id."
        ::= { r2TrapsObjects 1 }

    r2AlarmDescription OBJECT-TYPE
        SYNTAX      DisplayString
        MAX-ACCESS  not-accessible
        STATUS      current
        DESCRIPTION
            "r2 alarm description."
        ::= { r2TrapsObjects 2 }

    r2AlarmTime OBJECT-TYPE
        SYNTAX      DisplayString
        MAX-ACCESS  not-accessible
        STATUS      current
        DESCRIPTION
            "r2 alarm Time."
        ::= { r2TrapsObjects 3 }

    r2AlarmSeverity OBJECT-TYPE
        SYNTAX      INTEGER {
                        cleared(0),
                        indeterminate(1),
                        critical(2),
                        major(3),
                        minor(4),
                        warning(5),
                        information(6)
        }
        MAX-ACCESS  not-accessible
        STATUS      current
```

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```
        DESCRIPTION
            "r2 alarm Time."
        ::= { r2TrapsObjects 4 }

r2TrapsE1AlarmTrap  NOTIFICATION-TYPE
    OBJECTS {
        r2AlarmTrunkGroupId,
        r2AlarmDescription,
        r2AlarmTime,
        r2AlarmSeverity
    }
    STATUS      current
    DESCRIPTION
        "This trap indicates that the status of the indicated R1
        alarm."
    ::= { r2Traps 2 }

END
```

9 Software Licensing Keys

7515 follows a simple license schema based on hardware and software package.
The 7515 software package (PAK file) contains all software licenses required for field operation.
Corresponding license keys are acquired by purchase of the software package or the hardware boards respectively.

10 Restrictions and Known Problems

10.1 Restrictions removed/solved with the Release

10.1.1.1 Corrections and improvements in Release S31B0110

Identifier	Title	Customer Reference
GW1_00166831	7515 MGW does it support FQDN in P-Preferred-ID	

10.1.1.2 Corrections and improvements in Release S31B0109

Identifier	Title	Customer Reference
GW1_00165495	data port down after 9000 call in exceptional case	

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Identifier	Title	Customer Reference
GW1_00165438	twp policys with the same trunk group id	
GW1_00165285	system crash after do register operation	
GW1_00164995	optimize the structure "SIPREG_TRKGRP_T"	
GW1_00166112	system crash after update new version	

10.1.1.3 Corrections and improvements in Release S31B0107

Identifier	Title	Customer Reference
GW1_00128004	R2: review the cpc and cdc mapping	BrT

10.1.1.4 Corrections and improvements in Release S31B0106

Identifier	Title	Customer Reference
GW1_00162543	Not register to standby server srdxhm	
GW1_00162399	set trunk group id error in policy	
GW1_00161668	system crash if sip stack is disabled	

10.1.1.5 Corrections and improvements in Release S31B0105

Identifier	Title	Customer Reference
GW1_00142358	R2: Dual Seizure may cause channel unavailable	
GW1_00142145	incorrect trunk group configuration cause GW crash	
GW1_00127995	UI: remove first mapped cdc/cpc causes crash	

10.1.1.6 Corrections and improvements in Release S31B0104

Identifier	Title	Customer Reference
GW1_00153229	system crash if disable sip stack	

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Identifier	Title	Customer Reference
GW1_00143195	Change calling number type of NR in dial plan NOK	
GW1_00142742	R2: context lock when DSP resource unavailable	
GW1_00152161	GW crash if peer reboot/SWO during traffic	
GW1_00143814	GW rebooted for unknown reason with foc file	
GW1_00154948	R2: tonet call fail if no defult PUID provisioned	
GW1_00154666	display name in To header is not correct	
GW1_00153039	T1 link format is wrong in trunk group conf	
GW1_00142890	FSB reboot in G711 ISDN load testing	
GW1_00142274	register failed but state is IS-REG	
GW1_00141320	R2: "vi trunk group conf" result empty on STB FSB	
GW1_00127978	R2 tonet: 200 OK/INVITE after 200 OK/PRACK nok	
GW1_00160445	system crash after received INVITE message	

10.1.1.7 Corrections and improvements in Release S31B0103

Identifier	Title	Customer Reference
GW1_00143064	use domain name in From, To, Request-URI	
GW1_00141325	channel 16 not used in T1 mode	
GW1_00141223	system crash using "def red ac 1 1 fsb"	
GW1_00140732	UIs for CAS should be unavailable for ISDN	
GW1_00143816	can not handle second 180 without SDP correctly	
GW1_00143745	coco/sip context lock after abnormal PRI scenario	
GW1_00142993	CMCC: remove unsupport a=alt line in SDP	
GW1_00142877	context lock when DSP resource unavailable (tonet)	

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Identifier	Title	Customer Reference
GW1_00142529	basic call failed and system crash	
GW1_00141535	remove dom dial plan entry not correct	
GW1_00141325	channel 16 not used in T1 mode	
GW1_00141253	R2: incorrect of channel16 creation	
GW1_00141033	R2 modify tkg config, CPC need syntax check	
GW1_00140735	Dial Plan become unavailable for unknown reason	
GW1_00138283	PRA DL always "established" if peer disable link	

10.1.1.8 Corrections and improvements in Release S31B0102

Identifier	Title	Customer Reference
GW1_00141322	R2: "def cas config" on ACT not synchronize on STB	
GW1_00140666	trunk group can not be deleted if no link is in	
GW1_00127938	Collect call is not released when B disconnects	
GW1_00141597	assign link to trunk group 255 unsucessfully	
GW1_00141013	system crash after dis/ena SIP stack state	
GW1_00142055	register to FS5000 failed	
GW1_00142054	received second 180 with SDP NOK	
GW1_00141724	register message cannot send out sometimes	
GW1_00141720	fax control mode is not correct when MGC control	
GW1_00141538	coco/dsp resource hang up after reboot FSB in PRI	
GW1_00127949	Q.850 reasons for E1 failure and no channel	
GW1_00142247	scm crash when enable-reg	
GW1_00142334	CMCC deregister flow failed if receive 401	

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Identifier	Title	Customer Reference
GW1_00142012	R2: cyclic allocate no work if only 1 channel free	
GW1_00141832	R2: trunks change to PRA cannot be free	
GW1_00141223	system crash using "def red ac 1 1 fsb"	
GW1_00140528	R2 incoming trunk not free when view but works	
GW1_00127934	IOT 037 - wrong cause if reanswer timer expires R2	

10.2 Restrictions

10.2.1 Configure Restrictions

10.2.1.1 Switch Type of E1/T1

E1/T1 cares the switch type. E1 supports follow switch type:
NET5, NET5SW.
T1 supports follow switch type:
5ESS，DMS100，4ESS，NI2，NTT

10.2.1.2 TDM type

- A7515 supports R2 or PRI in TDM side but they **can not** coexist in one 7515
- A7515 ISDN stack supports Qsig and PRI and they **can** coexist in one 7515

10.2.1.3 SW Upgrade/Downgrade

There are two public releases for 7515S, one is 2.0 release, the other is 3.1 release .
Normally , different SW version which belongs to the same release can be upgrade/Downgrade freely
But for the different release, the below restraint must be followed

Case 1:
Upgrade SW form 2.0 release to 3.1 with 2.0 config.txt
- 2.0 Features can be supported by small change of configuration
- 3.1 Features is not committed to be supported

Case 2:

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Upgrade SW form 2.0 release to 3.1 with the new config.txt of 3.1s
- 2.0 Features can be supported by small change of configuration
- 3.1 Features can be supported be supported

Case 3 :
Downgrade SW form 3.1 release to 2.0 release is not supported

10.2.1.4 Length of Called Number

The max length of called number that dial plan can support is 28. But the max length of called number that ISDN stack support is not 28, it depends on switch type.
So it is possible that call failed because the length of called number is larger than ISDN stack supported.

10.2.1.5 Basic call in OVERLAP mode

- Current version only supports overlap call in case isdn type is E1, for T1, overlap call not support
- Overlap feature is partly implemented , Overlap function described in TS183036 Version 0.10.1 (2007-11)is not fully complied

10.2.1.6 FAX

The following scenarios for fax are supported
- Signal-control mode T38 and T30
- Auto-Switch mode T30

The following scenarios for fax are partly supported
- Auto-Switch mode T38
  Fax call will be OK Only in two 7515 S GW

10.2.1.7 Abnormal SIP message

7515 has no response for the follow case:
- Invite message without mandatory header, such as call ID.
- Invite message with bad syntax of P-DCS-Billing-Info.

10.2.1.8 Network failure

Two modes are defined in 7515.
- RTCP packet loss.
- DSP Statistics.

But “DSP Statistics” is only supported in Fax.

10.2.2 Feature not Support

Follow features are not supported:
- HTTP authentication

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- Network Management
- Hitless Redundancy

10.2.3 Known Bugs/Workarounds

TBD

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